REMARKS

By this Amendment the specification has been amended to improve its presentation, claim 1 has been amended to include the features of claim 3 (now canceled) and otherwise improved, claims 2 and 4-8 have been amended to better define the intended subject matter, and new claims 9 and 10 have been added to define further features of the invention (see the specification at page 5, lines 3-4 and page 6, lines 18-19 relative to new claim 9, and page 5, lines 29-32 relative to new claim 10). Entry is requested.

In the outstanding Office Action the examiner rejected claims 1-5 under 35 U.S.C. 102(e) as being anticipated by Svean et al., and he objected to claims 6-8 as being based on a rejected base claim.

The inventors assert that amended claim 1 is patentable and that all of claims 1, 2 and 4-10 should be allowed.

The present invention is directed to compensating for body-conducted sound from a person's own voice ('occlusion effect', cf. e.g., page 1, lines 13-16). This is achieved by the inclusion of a feedback noise control system that analyzes the sound in the cavity (which mainly originates from the person's own voice) to attenuate the low frequency part of the signal that is presented to the user via the receiver. The control system works in the analogue domain to avoid delays which are incurred by A/D and D/A conversions.

On the other hand, Svean et al. aims at compensating the air-borne sound from a person's own voice (which becomes distorted and/or attenuated because of the earplug). This is achieved by picking up the sound field produced. by the person's own voice in the cavity between the earplug and the tympanic membrane. This signal is amplified (E4) and digitized (A/D-converter E5) and processed in a DSP, where it is analyzed and classified to be able to select a filter H1(f), H2(f) and H3(f) appropriate for vowel sounds, nasal sounds and fricative sounds, respectively. The processed signal sound is presented to the user after D/A-conversion via a receiver (SG) (reference is made to col. 7, line 26 - col. 8, line 17, Figs. 1-4). From col. 8, lines 18-41, describing how the filter characteristics, analysis and classification can be derived from experiment, it becomes clear that the frequency range considered is from 100 Hz to 1400 Hz and that 'The transfer functions of the filters described in connection with Fig. 4 (H1-H3) may be based on diagrams H(f), the spectral density levels of the free field microphone M3 subtracted from the corresponding levels of the omitted staff low frequency part of the signal picked up in the cavity between the earpiece and the tympanic membrane as defined by the present invention. Further, the processing in Svean et al. of the signal from the 'inner microphone' (M2) is performed in the digital domain (which additionally would make it too

slow for compensating for the body-conducted sound from a person's own voice ('occlusion effect').

It is thus respectfully submitted that Svean et al. does not teach or suggest the present invention as defined in claim 1 or the claims dependent thereon.

Favorable reconsideration is requested.

A supplemental page 11 for this application containing an abstract of the disclosure is submitted herewith.

Respectfully submitted,

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